## CLAIMS

1. A waveform quality measurement apparatus, comprising:

means for providing a plurality of offsets of parameters of an actual signal with respect to an ideal signal;

means for compensating the actual signal with the plurality of offsets to generate a compensated actual signal;

means for filtering the compensated actual signal to generate a filtered signal;

means for modifying the ideal signal to correspond to the filtered signal to generate a modified signal; and

means for determining the waveform quality measurement in accordance with the modified ideal signal and the filtered signal.

2. The apparatus of claim 1, wherein the means for providing a plurality of offsets comprises means for providing a frequency offset, a time offset, and a phase offset.

3. The apparatus of claim 1, wherein the means for compensating the actual signal with the plurality of offsets comprises means for compensating in accordance with the following equation:

$$y(t) = x(t - \hat{\tau}_0)e^{j[\Delta\hat{\omega} \cdot t + \hat{\theta}_0]}$$

where:

$y(t)$ is the compensated actual signal;

$x(t)$ is the actual signal;

t is time;

j is an imaginary unit;

$\Delta\hat{\omega}$ is the frequency offset;

$\hat{\tau}_0$ is the time offset; and

$\hat{\theta}_0$ is the phase offset.

4. The apparatus of claim 1, wherein the means for filtering comprises means for assigning the compensated actual signal a value that is zero in intervals to be filtered and non-zero elsewhere.

5. The apparatus of claim 4, wherein the means for filtering comprises means for assigning the compensated actual signal a value that is non-zero over an elementary unit of the actual signal.

6. The apparatus of claim 4, wherein the means for assigning the compensated actual signal value comprises:

means for defining a function with a value that is zero in intervals to be filtered and non-zero elsewhere; and

means for multiplying the compensated actual signal by the function.

7. The apparatus of claim 6, wherein the means for defining a function comprises means for defining a function with a value that is non-zero over an elementary unit of the actual signal.

8. The apparatus of claim 1, wherein the means for modifying the ideal signal comprises means for generating the modified ideal signal to have a value that is zero in intervals where the filtered signal has a value of zero and non-zero elsewhere.

9. The apparatus of claim 1, wherein the means for modifying the ideal signal comprises means for assigning the ideal signal a value that is zero in intervals where the filtered signal has a value of zero and non-zero elsewhere.

10. The apparatus of claim 9, wherein the means for assigning the ideal signal a value comprises:

means for defining a function with a value that is zero in intervals where the filtered signal has a value of zero and non-zero elsewhere; and

means for multiplying the compensated actual signal by the function.

11. The apparatus of claim 5, wherein the means for determining the waveform quality comprises means for calculating a first overall modulation accuracy.

12. The apparatus of claim 11, wherein the means for calculating a first modulation accuracy comprises means for calculating in accordance with the following equation:

$$\rho_{\text{overall-1}} = \frac{N \cdot \sum_{j=1}^{N} \left| \sum_{k=1}^{M} Z_{j,k} R_{j,k}^{*} \right|^2}{\left\{ \sum_{j=1}^{N} \sum_{k=1}^{M} \left| R_{j,k} \right|^2 \right\} \cdot \left\{ \sum_{j=1}^{N} \sum_{k=1}^{M} \left| Z_{j,k} \right|^2 \right\}}$$

where:

$\rho_{overall-1}$ is the first overall modulation accuracy;

j is an index designating an elementary unit of signals;

N is a summation limit designating a number of elementary units;

k is an index designating a sample in the elementary unit;

M is a summation limit designating a number of samples in the elementary unit;

$Z_{j,k}$ = z[M(j-1)+k] is a $k_{th}$ sample in the $j_{th}$ elementary unit of the filtered signal; and

$R_{j,k}$ = r[M(j-1)+k] is a $k_{th}$ sample in the $j_{th}$ elementary unit of the ideal signal.

13. The apparatus of claim 11, further comprising means for calculating a second overall modulation accuracy.

14. The apparatus of claim 13, wherein the means for calculating a second modulation accuracy comprises means for calculating in accordance with the following equation:

$$\rho_{\text{overall-2}} = \frac{N \cdot \sum_{j=1}^{N} \left| \sum_{k=\frac{M}{2}+1}^{M+\frac{M}{2}+1} Z_{j,k} R_{j,k}^{*} \right|^2}{\left\{ \sum_{j=1}^{N} \sum_{k=\frac{M}{2}+1}^{M+\frac{M}{2}+1} \left| R_{j,k} \right|^2 \right\} \cdot \left\{ \sum_{j=1}^{N} \sum_{k=\frac{M}{2}+1}^{M+\frac{M}{2}+1} \left| Z_{j,k} \right|^2 \right\}}$$

where:

$\rho_{overall-2}$ is the second modulation accuracy;

j is an index designating an elementary unit of signals;

N is a summation limit designating a number of elementary units;

k is an index designating a sample in the elementary unit;

M is a summation limit designating a number of samples in the elementary unit;

$Z_{j,k} = z[(M + \frac{M}{2} + 1) \cdot (j-1) + k]$ is a $k_{th}$ sample in the $j_{th}$ elementary unit of the filtered signal; and

$R_{j,k} = r[(M + \frac{M}{2} + 1) \cdot (j-1) + k]$ is a $k_{th}$ sample in the $j_{th}$ elementary unit of the ideal signal.

15. The apparatus of claim 4, wherein the means for determining the waveform quality comprises means for calculating a modulation accuracy for a time division channel.

16. The apparatus of claim 15, wherein the means for calculating a modulation accuracy for a time division channel comprises means for calculating in accordance with the following equation:

$$\rho_{\mathrm{TDM\_channel}} = \frac{N \cdot \sum_{j=1}^{N} \left| \sum_{k=1}^{M} Z_{j,k} R_{j,k}^{*} \right|^{2}}{\left\{ \sum_{j=1}^{N} \sum_{k=1}^{M} \left| R_{j,k} \right|^{2} \right\} \cdot \left\{ \sum_{j=1}^{N} \sum_{k=1}^{M} \left| Z_{j,k} \right|^{2} \right\}}$$

where:

$\rho_{TDM\_channel}$ is the modulation accuracy for the time division channel *TDM_channel*;

j is an index designating an elementary unit of signals;

N is a summation limit designating a number of elementary units;

k is an index designating a sample in the elementary unit;

M is a summation limit designating a number of samples in the elementary unit;

$Z_{j,k} = z[M(j-1)+k]$ is a $k_{th}$ sample in the $j_{th}$ elementary unit of the filtered signal; and

$R_{j,k} = r[M(j-1)+k]$ is a $k_{th}$ sample in the $j_{th}$ elementary unit of the ideal signal.

17. The apparatus of claim 4, wherein the means for determining the waveform quality measurement comprises means for calculating code domain power coefficients.

18. The apparatus of claim 17, wherein the means for calculating code domain power coefficients comprises means for calculating in accordance with the following equation:

$$\rho_{TDM\_channel,i} = \frac{N \cdot \sum_{j=1}^{N} \left| \sum_{k=1}^{M} Z_{j,k} R'^{*}_{i,j,k} \right|^2}{\left\{ \sum_{j=1}^{N} \sum_{k=1}^{M} \left| R'_{i,j,k} \right|^2 \right\} \cdot \left\{ \sum_{j=1}^{N} \sum_{k=1}^{M} \left| Z_{j,k} \right|^2 \right\}}, \quad i = w_1, \ldots, w_v$$

where:

$\rho_{TDM\_channel,i}$ is the code domain coefficient for a time division channel *TDM_channel* and a code channel *i*;

w1 is a first code channel for the time division channel *TDM_channel*;

wv is a last code channel for time division channel *TDM_channel*;

j is an index designating an elementary unit of signals;

N is a summation limit designating a number of elementary units;

k is an index designating a sample in the elementary unit;

M is a summation limit designating a number of samples in the elementary unit;

$Z_{j,k} = z[M(j-1)+k]$ is a kth sample in the jth elementary unit of the filtered signal; and

$R'_{i,j,k} = R'_i\left[M(j-1)+k\right]$ is a kth sample in the jth elementary unit of the i-th code channel of the ideal signal.